Evaluation of a QoS Support Mechanism in Unified Network Architectures

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Rezumat. *Creșterea cererilor de servicii de bandă largă a dus, pentru diverse tehnologii de acces, la implementarea mecanismelor de asigurarea a calității serviciului pentru a se oferi performanțe ridicate prin prioritizarea traficului. Cu toate acestea, în arhitecturi de rețele unificate ce integrează multiple tehnologii de acces, nu se poate garanta calitatea serviciului pentru o transmisie cap-la-cap. Această problemă ar putea fi rezolvată dacă ruta ar fi aleasă pe baza cerințelor aplicației. Prin urmare, lucrarea de față propune un mecanism de asigurarea a suportului QoS care selectează o rută convenabilă și prezintă o analiză comparativă a performanțelor acestuia cu cele obținute utilizând suportul QoS existent în sistem.*

Cuvinte cheie: *QoS, QoE, rețele unificate.*

Abstract. *The increasing demand for high capacity services in communication networks has led to the implementation, in certain network technologies, of different QoS support mechanisms that would prioritize the traffic and provide high performances. However, when integrating multiple technologies in a unified network architecture, the end-to-end QoS support cannot be guaranteed. This problem may be solved if the end-to-end path is selected considering the application requirements. Therefore, this paper proposes a QoS support mechanism that selects a suitable end-to-end path and compares its performances to the ones provided by the existing QoS support.*

Keywords: *QoS, QoE, unified networks.*

1. INTRODUCTION

The advances in communications networking aim at providing better performances for the traffic while optimizing the resource management.

The concept of Quality of Service (QoS) refers to the applications that have certain requirements in terms of bandwidth, end-to-end delay or jitter. More than that, the quality of a transmission is also evaluated from the user's point of view. Thus, the concept of Quality of Experience (QoE) was introduced to reflect the user's opinion towards the quality of a service. However, the evaluation of QoE is more difficult because it involves non-technical factors like psychological or social factors *[1].*

Nevertheless, the rapid development of wireless networks and user mobility presents more challenges for providing quality of service. Also, multimedia applications are types of applications that have stringent requirements: high bandwidth and reduced delay and jitter *[2].* To provide quality of service several mechanisms were implemented.

Nowadays, a telecommunications system may include several networks, administrative domains

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and access technology. Although there are mechanisms that provide quality of service for intra-domains, when using a unified network, ensuring end-to-end quality of service for an inter-domain application becomes an issue *[3]-[6].*

Therefore, this paper proposes and evaluates a profile-based QoS support mechanism that provides an end-to-end quality of service for applications running in unified network architectures. The remaining of this paper is organized as follows: *Section 2* presents the concepts of QoS and QoE, *Section 3* briefly presents the existing QoS mechanisms implemented in the IEEE 802.11 *[7], [8]* and IEEE 802.16 *[9], [10]* Standards, in *Section 4* the proposed mechanism is described and its performances are evaluated in *Section 5*. Finally, *Section 6* concludes the paper.

2. QUALITY OF SERVICE AND QUALITY OF EXPERIENCE

2.1. Quality of Service

Several definitions of Quality of Service are presented in literature. ITU-T defines this concept in Rec. E.800 as "*the ability of a network or network portion to provide the functions related to communications between users*" *[11]* and IETF defines it as "*a set of service requirements to be met by the network while transporting a flow*" *[12].*

QoS may be evaluated by means of different parameters. These parameters may vary according to the services that are played. Regarding the performances of the network, the most popular QoS parameters are:

The delay, which represents the time spent by a packet to reach the destination. Four types of delay are cumulated in this parameter: processing delay (the time needed by the network entities to process the packet), queuing delay (the time spent waiting in queues – this value increases in congested networks), transmission delay (the time spent when transmitting a packet at a certain bit rate), and propagation delay (the amount of time needed for the signal propagation). For a certain path, the processing, propagation and transmission delays do not change. The variation of the end-to-end delay is due to the queuing delay.

The jitter represents the variation of the delay and can be compensated using buffers.

The bandwidth is the capacity of a segment or an end-to-end path. It is measured in *bps*.

Packet loss represents the packets that fail to reach the destination. The packet loss has two main causes: packet errors due to the poor connection quality and packet drops due to the congestion.

Another concept related to the QoS is the class of service. Some classes are defined by quantitative or qualitative parameters. Usually there are certain imposed value ranges for the parameters describing a certain service class.

For VoIP applications, the critical parameter is the delay. In *[12]* a one-way delay of 150 ms is considered acceptable, while a roundtrip latency of 250 ms can be detected by the end-user. The jitter can also be annoying, but it can be resolved with buffers if it is less than 100ms. Moreover, VoIP applications can barely tolerate a packet loss rate greater than 1%.

Regarding video transmissions, *[13]* recommends a latency value lower than 4-5 s and packet loss lower that 5% for the streaming-video application and a latency lower than 150 ms, jitter lower than 30 ms and under 1% packet loss for the interactive video application.

2.2 Quality of Experience

Quality of Experience (QoE) is defined by the ITU-T *[14]* as *"the overall acceptability of an application* *or service, as perceived subjectively by the end-user"* meaning that the QoE measures the user's expectations towards the quality of a service. The quality of a transmission is affected by factors like the quality of the source, the impairments introduced by the network, protocols, codecs and terminals.

Although when determining the QoE certain QoS parameters are taken into account, the user perception is also influenced by non-technical factors like the environment and social and psychological factors. Therefore, the QoE augments the QoS by including the user's expectations regarding the performances of a transmission.

Because it reflects the user's opinion, the QoE cannot be easily evaluated. There are two main approaches when trying to determine the QoE: a subjective method and an objective one.

The subjective method makes use of live testing and experiments including people. The participants evaluate the quality of a certain service by means of a score on a pre-established scale that describes the amount of user satisfaction. The final result includes the impact of non-technical factors but the method itself is expensive and time consuming *[1].*

The objective method requires the development of a new metric based on a mathematical model that associates the network performances to the user's perception. The main three steps that need to be considered when developing such a model *[1]* are the following:

1) the analysis of the QoS parameters that have a great impact on the perceived quality (i.e. delay, jitter, packet loss *[15]*) and determining their effect on the QoE;

2) measuring the previously determined parameters;

3) using mapping metrics to determine the degree of user satisfaction based on the measured parameters.

2.2.1. Objective QoE analysis: the E-model and Mean Opinion Score

The E-model *[16]* is the most popular objective method used to estimate the quality of an end-to-end real-time transmission (especially voice) and to anticipate the user's experience taking into account certain impairment factors (i.e. delay and packet loss). The primary output of this model is the R factor, given in *(1)*:

$$`R = R_0 - I_s - I_d - I_{e-eff} + A \qquad (1)$$

R_0 is the Signal to Noise Ratio, I_s represents the impairments that occur at the same time as the voice signal, I_d represents the impairments caused by the delay, $I_{e\text{-}eff}$ is the effective equipment impairment factor (caused by the low rate codecs), and *A* is the advantage factor, which may compensate for some of the impairment factors assuming that there are other advantages of access to the user *[16].*

Processing the R factor may provide an estimation of the user's opinion. Some QoE metrics derived from the R factor are Good or Better (GoB), Poor or Worse (PoW) and Mean Opinion Score (MOS).

The Mean Opinion Score is the most popular QoE metric and is defined in *[16]* as *"the value on a predefined scale that a subject assigns to his opinion of the performance of the telephone transmission system used either for conversation or only for listening to spoken material"*.

The MOS is a value between 1 and 5, from the lowest to the highest perceived quality and is also used to evaluate voice and video transmissions. The mapping between the absolute value of MOS, the perceived quality descriptor and the degradation of the transmission from the user's point of view is presented in *Table 1*.

Table 1

MOS mapping table.

MOS	Quality descriptor	Degradation
5	Excellent	Imperceptible
4	Good	Perceptible
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

The correspondence between the R factor and the MOS is given in (2)

$$\begin{gathered} R < 0, \qquad MOS = 1 \\ 0 < R < 100, \\ MOS = 1 + 0.035R + \\ + R(R-60)(100-R)7 \cdot 10^{-6} \\ R > 100, \qquad MOS = 4.5 \end{gathered} \tag{2}$$

3. EXISTING QOS SUPPORT IN WIRELESS ACCESS TECHNOLOGIES

3.1. QoS in the IEEE 802.11 Standard

IEEE defines eight types of traffic. They are mapped onto the user priorities (UP) used to differentiate the quality of service. For the wireless networks, IEEE defines four types of Access Categories (AC) to differentiate the services. The correspondence between ACs, UPs and traffic types is presented in *Table 2 [8].*

Table 2

Mapping ACs, UPs and traffic types

AC	UP	Traffic type
0 (BK)	1	Background
	2	
1 (BE)	0	Best Effort
	3	
2 (VI)	4	Video
	5	
3 (VO)	6	Voice
	7	

The IEEE 802.11b (now included in the revised IEEE 802.11 Std. *[7]*) is the most popular and widely implemented Wireless Local Area Network (WLAN) technology *[20].* There are two basic modes of operation in WLAN: Distributed Coordination Function (DCF) and Point Coordination Function (PCF).

The DCF mode allows the stations to access the medium without a centralized control, using the CSMA/CA mechanism. The PCF mechanism requires a point coordinator (PC) that allows the stations to transmit without contention for the wireless medium. Although the PCF mechanism prioritizes the traffic, none of these mechanisms offer QoS support because there is no differentiation between the services.

The IEEE 802.11e amendment *[8]* was implemented to solve this problem. Thus, the Hybrid Coordination Function (HCF) was introduced and two new medium access mechanisms were implemented: Enhanced Distributed Channel Access (EDCA) and HCF Controlled Channel Access (HCCA).

EDCA provides a differentiated and distributed access to the channel resources and implements eight priority levels (0 to 7) assigned to the four access categories (BK, BE, VI, VO). Every frame that reaches the MAC layer is mapped on the corresponding category according to its priority. Every AC is characterized by parameters like Contention Window (CW) or Transmission Opportunity (TXOP). The higher the priority, the smaller the CW duration. TXOP is defined as the period of time when a station may initiate the transmission without contenting for the channel.

The HCCA mechanism uses a central entity named Hybrid Coordinator (HC) that controls the medium access. Each station that needs to access the resources has to send a reservation request to the HC. In infrastructure association mode, the HC is the Access Point (AP). Each station with QoS support may transmit packets only in the previously allotted TXOP.

3.2. QoS in the IEEE 802.16 Standard

Applications like multimedia services are hard to manage in traditional networks due to the small throughput and reduced accessibility. Unlike cellular networks that have reduced capacity or the WLANs that provide reduced coverage, WiMAX (Worldwide Interoperability for Microwave Access) that implements the IEEE 802.16 Std. *[9], [10]* provides wider bandwidths and roaming for mobile users.

The QoS support for wireless networks is usually provided at the MAC layer *[17].* At the MAC layer of the IEEE 802.16 metropolitan networks, five types of traffic scheduling services are defined:

– unsolicited grant service (UGS) – services that require certain delay performances and send equal packets at equal intervals of time.

– extended real-time polling service (ertPS) – traffic resembling the UGS but with variable packet size.

– real-time polling service (rtPS) – data streams that send packets of variable size at equal intervals of time.

– non-real-time polling service (nrtPS) – data streams with variable packet rate

– best effort service (BE) – traffic that does not have strict performance requirements and is delivered according to the available resources.

Table 3 presents the main characteristics of these QoS services.

In the existing network technologies, the services are prioritized, classified, scheduled and delivered according to the QoS support implemented in that specific technology. However, when using unified network architectures (that include several types of networks and access technologies) the end-to-end quality of the transmission cannot be provided because there are certain compatibility issues between the existing intra-domain QoS mechanisms. Therefore, the applications are delivered in a best-effort manner.

The QoS support mechanism proposed in this paper is meant to solve this problem and provide an end-to-end QoS support in heterogeneous networks. The mechanism is tested and evaluated using network simulations.

Table 3

Scheduling services in IEEE 802.16

QoS service	Packet dimension	Delay requirements	Bandwidth request	Application
UGS	constant	Strict	Necessary	T1/E1, VoIP
ertPS	variable	Strict	Necessary	VoIP with silence suppression
rtPS	variable	Not so strict	Unicast, piggyback	MPEG Stream
nrtPS	variable	Not specified	Unicast, piggyback contention	FTP
BE	variable	Not specified	Unicast, piggyback contention	WWW applications

4. PROFILED-BASED QOS SUPPORT

To provide QoS support in unified networks, the proposed mechanism intends to accommodate the requirements of the application to the network context. This is done by determining the critical parameters for the application that is to be run and then selecting the proper end-to-end path.

This algorithm uses three types of QoS profiles: QoS Requested Profile, QoS Path Profile and QoS Available Profile.

The profile-based mechanism identifies the specific application requirements and uses the QoS Requested Profile to specify the critical parameter for the transmission (either delay or jitter). The QoS Path Profile includes the best end-to-end path that was determined in the probing phase of the algorithm and the cumulative and average values of the measured QoS parameters on that path.

The probe messages mimic the application (same packet size and interval between packets) and include specific descriptors for the Path and Requested Profiles.

The probes are transmitted in a hop-by-hop manner to the adjacent nodes, starting with the source node and ending on the destination node. An intermediate node processes each complete probe set. The average values of the critical parameters are computed and the set of probes providing the best performances is forwarded to the adjacent nodes. At a certain point, the set of probes with the best performances ought to reach the destination. At the destination node, the process is finalized by selecting the end-to-end path that provides the best performances in the network.

The determined end-to-end path is then included in the QoS Available Profile and an answer is sent back to the source. This message informs the source that the selected route is the one providing the best performances. On its way back to the source, the message labels the nodes in the path and marks the path for the application.

4.1. Network probing

To determine the network state, the proposed mechanism performs the probing of the network. During this phase, the mechanism sends probe messages in the network to determine, based on the critical parameters, an end-to-end path that would satisfy the application requirements.

The number of probes in a set determines the network overload and the time needed to determine the best end-to-end path. Therefore, in the calibration phase, we have conducted several simulations to determine the optimum number of probes.

The results showed that the time needed to determine the best path is proportional to the number of probes in a set but is not drastically influenced by the critical parameter.

If a small number of probes is used, the estimated performances are not accurate and therefore, a path that does not provide the best performances is selected.

Increasing the number of probes does increase the prediction accuracy, but a high number of probes in a set may increase the network overload and cause congestions.

The simulations also showed that from a certain point, increasing the number of probes in a set does not improve the accuracy of the estimated performances compared to the real ones.

Therefore, there is a number of probes in a set that guarantees the selection of the end-to-end path that estimates most accurately the jitter and the delay of the transmission, without packet loss.

We have empirically determined that the optimum number of probes in a set is determined by the interval between packets that characterizes the application. The rule is given in (3):

$$Number\ of\ probes\ per\ set = \\ = \frac{1}{Interval\ between\ packets\ (s)} \quad (3)$$

5. EVALUATION OF THE PROPOSED SCENARIO

5.1. The network simulator

To evaluate the performances of the proposed mechanism we have created a suitable integrated network and simulated its behavior using QualNet Developer 5.1.

QualNet Developer 5.1 is a network simulator developed by Scalable Network Technologies (SNT) that allows protocol implementation, configuration and simulation of network topologies and analysis of network performances *[18].*

QualNet Developer 5.1 provides some model libraries implementing several standard specifications, such as: cellular model, UMTS, WLAN, WiMAX, or ZigBee. Moreover, the source code is available to the user and additional protocols may be implemented and included in the application.

The proposed mechanism was implemented in C++ and included in QualNet Developer 5.1 as a patch.

5.2. Scenario description

To evaluate the proposed mechanism we have created a simulation scenario that unifies multiple access technologies. As *Figure 1* shows, the access networks at the source are connected to the access networks from the destination through an intermediate wired network (core network infrastructure).

The source node (SN) is placed in the coverage area of two different types of access technologies: a wireless local area network implementing the IEEE 802.11 Standard *[7]* and a wireless metropolitan network implementing the IEEE 802.16 Standard *[9]*. The same applies for the destination node (DN) that is in the coverage areas of two networks, as illustrated in *Figure 1*. Therefore, each of these nodes may access the resources through the access point (AP) or the base station (BS) in the corresponding access networks. The wired links interconnecting the routers (R) are characterized by different bandwidths and delays, providing several end-to-end paths with different performances. In each access network there are several users that generate background traffic in the network.

In order to provide a variable network context, traffic generators are used to run different types of applications in the wired network and in the source and destination access segments. *Table 4* presents the configuration parameters for the background traffic.

The performance evaluation is done for a test application that models a video transmission between the source node and the destination node. Considering that there are video codecs characterized by sending equal packets at equal intervals of time, a Video on Demand (VoD) stream is modeled in QualNet using a CBR application. The configuration of the parameters is as follows: ***packet dimension = = 1370 bytes, interval between packets = 3.26 ms, total number of transmitted packets = 32448, data rate = 3361 kbps, duration = 106 s.***

5.3. Simulation results

To emphasize the advantages of the proposed mechanism we compare its performances to the performances obtained when the existing QoS support is used for different routing protocols. The test application is evaluated in terms of end-to-end delay, average jitter and packet loss on the selected path. Also, the user's opinion is estimated using the previously presented MOS.

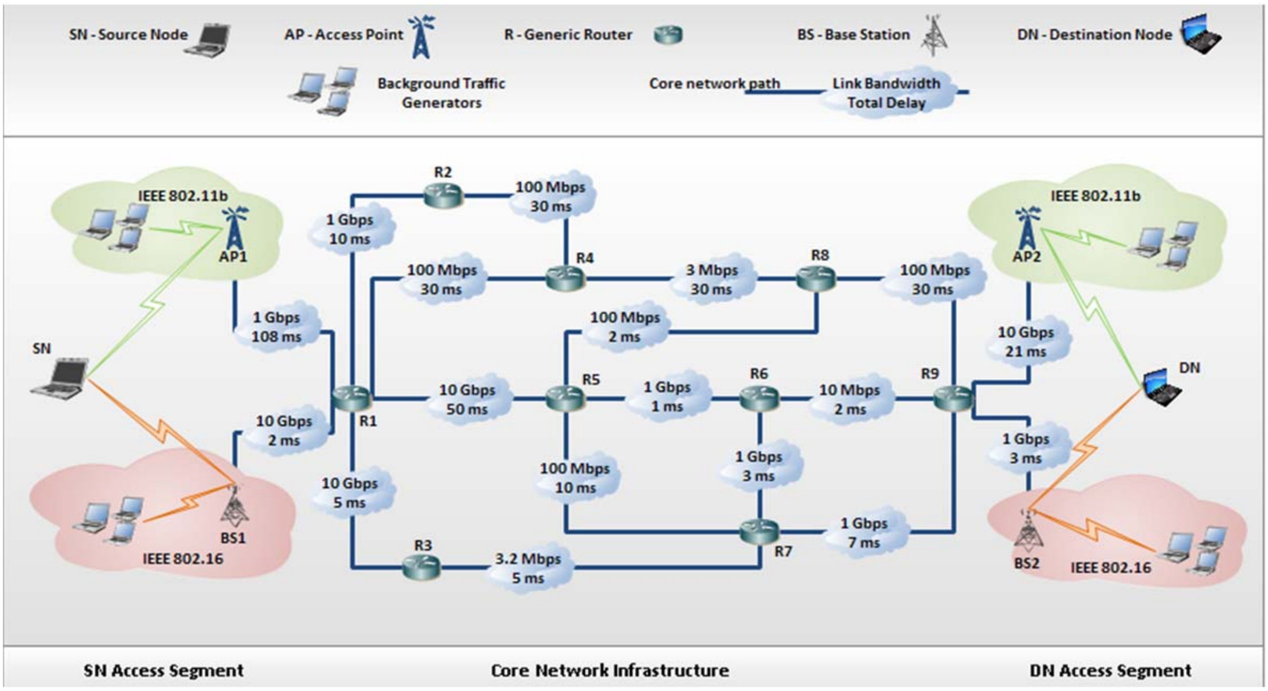

Fig. 1. Test scenario architecture.

Table 4

Background traffic configuration (SS - source segment, DS - destination segment)

Application	Parameters					
FTP	*Source Node*	*Destination Node*	*Packet dimension (bytes)*		*Start time (s)*	*duration (s)*
	22 (WLAN DS)	23 (WLAN DS)	1500		5	175
	35 (WLAN DS)	32 (WLAN DS)	1500		5	175
CBR	*Source Node*	*Destination Node*	*Packet dimension (bytes)*	*Interval between packets (ms)*	*Start time (s)*	*Duration (s)*
	16 (WLAN SS)	25 (WLAN DS)	1370	15.38	5	175
	31 (WLAN SS)	24 (WLAN DS)	1370	15.38	5	175
	R8	R4	1370	15.38	5	175
	R8	R9	1370	15.38	5	175
	R4	R8	1370	15.38	5	175
	R2	R4	1200	14.5	5	175
Traffic generator	*Source Node*	*Destination Node*	*Packet dimension (bytes)*	*Interval between packets (ms)*	*Start time (s)*	*Duration (s)*
	20 (WiMAX SS)	21 (WiMAX SS)	256	20	5	175
	14 (WLAN SS)	15 (WLAN SS)	256	20	5	175
	28 (WiMAX DS)	29 (WiMAX DS)	256	20	5	175

If there is no QoS mechanism available, the applications are delivered in a BE manner. When enabling the existing QoS support, the application is run according to the priority rules indicated by the IP precedence. Considering that the test application models VoD traffic, the IP precedence is set to 5. *Table 5* shows the performances of the existing QoS support.

The results show that the same end-to-end path is selected (SN-AP1-R1-R4-R8-R9-AP2-DN) regardless of the protocol.

Table 5

Performances of the existing QoS support

BE	Routing protocol	OSPFv2	RIP	Bellman Ford
	End-to-end path	SN-AP1-R1-R4-R8-R9-AP2-DN		
	Average delay (ms)	718.58	718.58	713.54
	Average jitter (ms)	1.60	1.60	1.63
	Packet loss rate (%)	29.24	29.24	29.31
	MOS	1	1	1
IP Precedence =5	Routing protocol	OSPFv2	RIP	Bellman Ford
	End-to-end path	SN-AP1-R1-R4-R8-R9-AP2-DN		
	Average delay (ms)	624.45	624.45	624.41
	Average jitter (ms)	1.34	1.34	1.34
	Packet loss rate (%)	14.44	14.44	14.44
	MOS	1	1	1

Although the performances are improved when the traffic prioritization is used, the end-to-end delay exceeds the limitations for real-time traffic *[2]*. Therefore, the estimated user's opinion indicates a bad quality of the transmission. Thus, the results show that there is no correlation between the selected end-to-end path and the application requirements and the mechanisms are not capable of delivering the required end-to-end QoS support.

According to *(6)*, for the proposed mechanism, the number of probes in a set is 307. *Table 6* presents the comparative results for the performances obtained when using the existing QoS support and the performances obtained with the proposed mechanism.

When enabling the profile-based QoS support, the average delay decreases and there is no packet loss on the selected path. Consequently, the MOS for the selected paths indicates a good to excellent transmission and an almost imperceptible degradation. These improvements are due to the fact that the proposed mechanism selects a path that respects the application requirements. *Figure 2* illustrates the selected end-to-end paths.

5.4. Evaluation of the reconfigured application

To improve further more the performances of the transmission an application reconfiguration may be required. The reconfigured test application is modeled with a CBR and the rate is decreased by increasing the interval between packets.

The reconfiguration of the parameters is as follows: ***packet dimension = 1370 bytes, interval between packets = 6.09 ms, total number of transmitted packets = 17385, data rate = 1799 kbps, duration = 106 s.***

The test scenario is the same and the simulation results for the reconfigured application when the existing QoS support is enabled are presented in *Table 7*.

Table 6

Performances of the proposed QoS support mechanism

Existing QoS support IP Precedence =5	Selected path	SN-AP1-R1-R4-R8-R9-AP2-DN
	Average delay (ms)	624.45
	Average jitter (ms)	1.34
	Packet loss rate (%)	14.44
	MOS	1
Profile-based QoS support Critical parameter = delay	Selected path	SN-BS1-R1-R5-R6-R9-BS2-DN
	Average delay (ms)	158.41
	Average jitter (ms)	5.60
	Packet loss rate (%)	0
	MOS	4.32
Profile-based QoS support Critical parameter = jitter	Selected path	SN-AP1-R1-R5-R6-R9-BS2-DN
	Average delay (ms)	185.69
	Average jitter (ms)	5.45
	Packet loss rate (%)	0
	MOS	4.28

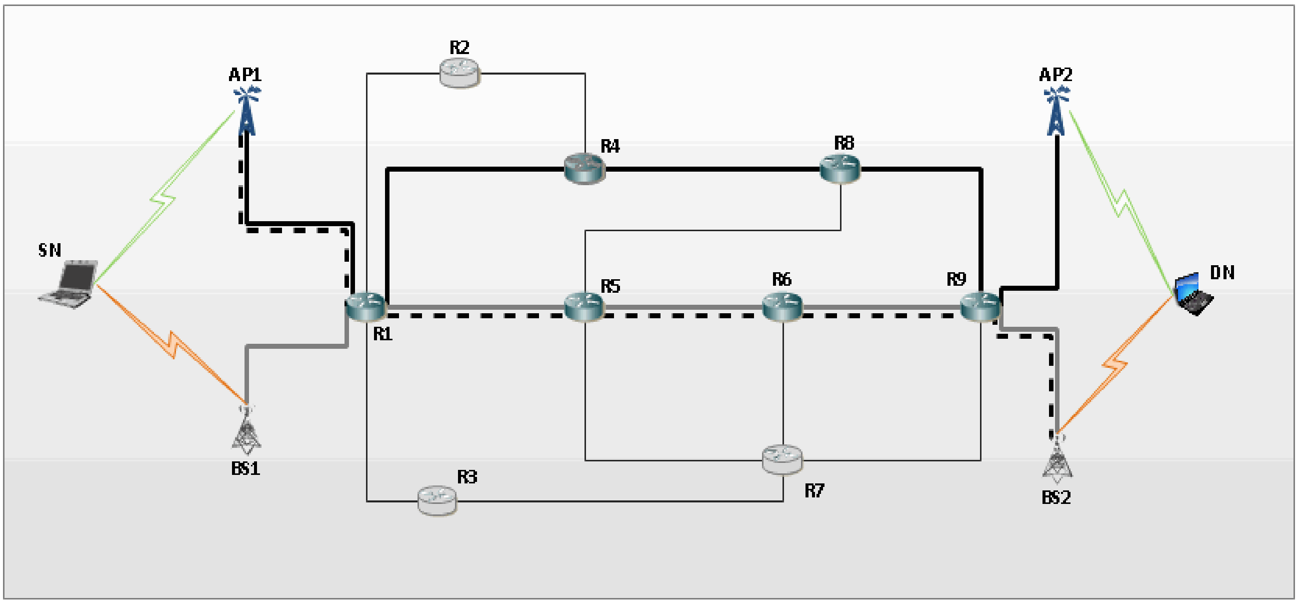

Fig. 2. Selected paths: **existing QoS support - solid black line**; proposed QoS mechanism for **critical parameter set to delay - solid grey line**, proposed QoS mechanism for **critical parameter set to jitter - dashed grey line**.

Table 7

The performances of the existing QoS support for the reconfigured application

BE	Routing protocol	OSPFv2	RIP	Bellman Ford
	End-to-end path	SN-AP1-R1-R4-R8-R9-AP2-DN		
	Average delay (ms)	704.17	704.17	704.51
	Average jitter (ms)	2.67	2.67	2.72
	Packet loss rate (%)	15.31	15.31	15.61
	MOS	1	1	1
IP Precedence =5	Routing protocol	OSPFv2	RIP	Bellman Ford
	End-to-end path	SN-AP1-R1-R4-R8-R9-AP2-DN		
	Average delay (ms)	227.85	227.85	227.9
	Average jitter (ms)	2.04	2.04	2.04
	Packet loss rate (%)	0	0	0
	MOS	4.1	4.1	4.1

As expected, increasing the interval between packets decreases the delay and the packet loss. The quality of the transmission improves when the traffic prioritization is indicated through an IP precedence of 5, but the selected path remains the same. Therefore, the existing QoS mechanisms do not take into account the application characteristics when selecting an end-to-end path.

The performances of the reconfigured application were tested for the profile-based mechanism too. According to *(6)* the number of probes in a set for a packetization interval of 6.09ms should be set to 165. The results of the simulations for the evaluation of the proposed mechanism for the reconfigured application are presented in *Table 8.*

The paths indicated in *Table 8* are selected according to the critical parameter and the evaluation of the QoS and QoE parameters indicates better performances in the network. These paths are different from the path selected by the existing QoS support and from the paths selected for the original application. Thus, it is proven that the proposed QoS mechanism takes into account the requirements of the application when selecting an end-to-end path and consequently, the quality of the transmission is improved.

Table 8

Simulation results for the reconfigured application when the proposed QoS mechanism is enabled

	Critical parameter = delay	**Critical parameter = jitter**
Number of probes per set	165	
Total number of probes in the network	5280	
Path estimation time (s)	7.67	
Selected path	SN-BS1-R1-R3-R7-R6-R9-BS2-DN	SN-AP1-R1-R3-R7-R9-BS2-DN
Average delay (ms)	122.64	152.13
Average jitter (ms)	9.2	8.46
Packet loss rate (%)	0	0
MOS	4.34	4.32

6. CONCLUSIONS

Different standards implement different types of QoS support mechanisms. In IEEE 802.11, the HCF and EDCA and HCCA mechanisms are implemented along with the usage of four access categories (BK, BE, VI, VO) mapped on the user's priorities. In IEEE 802.16, five types of traffic scheduling services are implemented: UGS, ertPS, rtPS, nrtPS, BE.

However, when multiple technologies are integrated in a unified network architecture, the QoS support for an end-to-end transmission cannot be guaranteed.

The proposed mechanism makes use of three profiles that describe the application and the network behavior when the application is run: QoS Requested Profile, QoS Path Profile, QoS Available Profile.

To select the best end-to-end path, the mechanism performs the probing of the network and determines the path that has the minimum cumulative value of the critical parameter.

This mechanism was implemented as a patch and included in QualNet Developer 5.1, a network simulator that allows the user to create and configure network scenarios and simulate their behavior. The proposed mechanism is evaluated through simulations.

The test scenario represents a unified network architecture including WLAN and WIMAX wireless access networks and a wired intermediate network. Because the existing mechanisms cannot provide a high quality end-to-end transmission, the QoS support is provided using the proposed mechanism.

The comparative analysis of the simulation results show that the performances of the transmission are increased when the profile-based QoS support is activated. This is due to the fact that the proposed mechanism takes into account the application requirements and finds a path in the network that provides the best performances of the critical parameter.

To improve even more the performances, the application is reconfigured and a new set of simulations is run. The results show that the proposed mechanism respects the application requirements and finds suitable end-to-end paths in the network. Therefore, the performances of the end-to-end transmission are increased.

The performances for the test application (VoD modeled with a CBR) show that the profile-based mechanism provides an effective QoS support for an end-to-end transmission in a unified network.

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